OpenSIPS and WebRTC

Peter Kelly / pkelly@sourcevox.com
Who we are

- VoIP and OpenSIPs software development and consultancy
- Based in UK
- Some of larger customers are
  - Localphone
    - Retail ITSP offering (VoIP accounts, apps, DIDs in UK, US, Europe, Worldwide)
    - Over 1,000,000 users
  - Magic Telecom
    - US Facilities based CLEC
  - Voxbeam
    - Wholesale, A-Z Termination, VoIP reseller
    - US CLEC
    - Terminate ~20,000,000 mins/week internationally
- We use
  - OpenSIPs
  - Asterisk
  - FreeSWITCH
  - RabbitMQ
  - Redis
  - Hadoop
  - Homer
  - Sangoma
Workshop Aims

• Demonstrate new Websocket (secure) module
• Demonstrate how a browser can now be a SIP UA with OpenSIPs as the proxy
• Demonstrate how a browser can talk to the regular PSTN
• Live(!) and Exciting(!!) realtime OpenSIPs configuration!
What is WebRTC

• New standard ratified by W3C

• Defines how browser to browser media sessions can be established (Video and Audio)

• As of 2016
  - Good support in Chrome and Firefox
  - Android browser support
  - iOS browser support.
WebRTC = Standards

- Standardisation on Audio/Video encryption with DTLS/SRTP
- Standardisation on media NAT traversal using ICE
WebRTC !≠ SIP

- Transport agnostic
- SIP can carry SDP transport
- In addition to WebRTC, browsers now support Websocket connections
- OpenSIPS now supports websocket connections
Result

Browser WebRTC
+ Browser Websocket support
+ OpenSIPS Websocket support
= audio and video calls from a browser, via OpenSIPS
• Browser has no access to host IP.

• SIP request sent up websocket with no useful routing information.

• OpenSIPs must use `fix_nated_register()` to send requests and responses to IP and Port received (1,2)
WebRTC NAT

- Same problem with INVITE and 200OK reply
- OpenSIPS must use fix_nated_contact() so UAS and UAC insert correct IP for routing SIP packet.
Relay media to PSTN

- RTPEngine converts “ICE” SDP to regular SDP and vice versa
- Traditional PSTN gateways can talk to WebRTC media
- Result is Browser to PSTN calls

OpenSIPS Summit 2016, Amsterdam
Conclusions

- WebRTC enables browser to be used as a SIP UA
- Easier for end users: No software to download, no proxy, username, transport, STUN, password, SSL certificate info needed
- WebRTC standard delivers secure voice and audio
- ICE negotiation removes any real need for NAT handling for media
- Easy to bolt onto an existing OpenSIPS registrar using WS and WSS modules.
- Easy to integrate into existing web apps. e.g. PRESENCE support, MESSAGE support.
Code samples

The code used in the presentation can be found at

• https://github.com/petekelly/opensips-summit-2016
Thank You...

Peter Kelly
pkelly@sourcevox.com